Session Initiation Protocol (SIP)

SIP, the Session Initiation Protocol, is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP was developed within the IETF MMUSIC (Multiparty Multimedia Session Control) working group, with work proceeding since September 1999 in the IETF SIP working group.

A number of standardization organizations and groups are using or considering SIP:

- IETF PINT working group
- 3GPP for third-generation wireless networks
- Softswitch Consortium
- IMTC and ETSI Tiphon are working on SIP-H.323 interworking
- PacketCable DCS (distributed call signaling) specification

[News] [Overview] [Where is SIP being discussed?] [Search] [What SIP extensions are being planned?] [Drafts] [SIP grammar] [Implementations] [SIP service providers] [Vendors] [Mailing list] [Public SIP Servers] [Papers] [Talks] [Related Drafts and Documents] [Draft History] [Frequently Asked Questions (FAQ)] [Port Assignments and DNS] [Compact headers] [SIP-related events] [SIP interoperability testing events] [Press Coverage] [Emergency Calling] [Non-Internet-related SIP sightings] [Status and Schedule] [Internships and Jobs] [Basic telephony features] [Other SIP sites]

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News

- January 19, 2005: *Session Initiation Protocol (SIP)-H.323 Interworking Requirements* approved as Informational
- December 21, 2004: *The Alternative Network Address Types Semantics (ANAT) for theSession Description Protocol (SDP) Grouping Framework* approved (Proposed Standard)
- December 1, 2004: *Usage of the Session Description Protocol (SDP) Alternative Network Address Types (ANAT) Semantics in the Session Initiation Protocol (SIP)* approved (Proposed Standard)
- November 11, 2004: *Transcoding Services Invocation in the Session Initiation Protocol (SIP) Using Third Party Call Control (3pcc)* approved (informational)
- November 11, 2004: *Input 3rd-Generation Partnership Project (3GPP) Release 5 requirements on the Session Initiation Protocol (SIP) approved (informational)*
- *November 1, 2004: Update to the Session Initiation Protocol (SIP) Preconditions Framework approved as Proposed Standard*
- *November 1, 2004: RFC 3903 (Session Initiation Protocol (SIP) Extension for Event State Publication) published*
- *November 1, 2004: RFC 3880 (Call Processing Language (CPL): A Language for User Control of Internet Telephony Services) published*
- *October 22, 2004: RFC 3911 (The Session Initiation Protocol (SIP) "Join" Header) published*
- *September 28, 2004: RFC 3893 (Session Initiation Protocol (SIP) Authenticated Identity Body (AIB) Format) published*
- *September 28, 2004: RFC 3892 (The Session Initiation Protocol (SIP) Referred-By Mechanism) published*
- *September 28, 2004: RFC 3891 (The Session Initiation Protocol (SIP) "Replaces" Header) published*
- *September 15, 2004: A Presence-based GEOPRIV Location Object Format approved as Proposed Standard.*
- *August 31, 2004: RFC 3840 (Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)) published.*
- *August 31, 2004: RFC 3841 (Caller Preferences for the Session Initiation Protocol (SIP)) published.*
- *August 31, 2004: RFC 3842 (A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)) published.*
- *August 31, 2004: RFC 3856 (A Presence Event Package for the Session Initiation Protocol (SIP)) published.*
- *August 31, 2004: RFC 3857 (A Watcher Information Event Template-Package for the Session Initiation Protocol (SIP)) published.*
- *August 31, 2004: RFC 3858 (An Extensible Markup Language (XML) Based Format for Watcher Information) published.*
- *August 27, 2004: Indication of Message Composition for Instant Messaging approved as Proposed Standard.*
- *August 17, 2004: The tel URI for Telephone Numbers approved as Proposed Standard.*
- *August 17, 2004: Session Timers in the Session Initiation Protocol (SIP) approved as Proposed Standard.*
- *July 26, 2004: The Early Session Disposition Type for the Session Initiation Protocol (SIP) approved as Proposed Standard and Early Media and Ringing Tone Generation in the Session Initiation Protocol approved as BCP.*
- *July 21, 2004: RFC 3853 (S/MIME Advanced Encryption Standard (AES) Requirement for the Session Initiation Protocol (SIP)) published.*
- *June 29, 2004: RFC 3824 (Using E.164 numbers with the Session Initiation Protocol (SIP))*

published.

- *June 28, 2004: The Internet Assigned Number Authority (IANA) Header Field Parameter Registry for the Session Initiation Protocol (SIP) and The Internet Assigned Number Authority (IANA) Universal Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP) approved as BCP.*
- *June 2, 2004: Session Initiation Protocol (SIP) Extension for Event State Publication approved as Proposed Standard.*
- *June 1, 2004: The Session Inititation Protocol (SIP) 'Join' Header approved as Proposed Standard.*
- *May 28, 2004: CPL: A Language for User Control of Internet Telephony Services approved as Proposed Standard.*
- *May 19, 2004: H.350 Directory Services approved as an Informational RFC.*
- *May 17, 2004: The SPIRITS (Services in PSTN requesting Internet services) Protocol approved as Proposed Standard.*
- *May 10, 2004: SIP Authenticated Identity Body (AIB) Format approved as Proposed Standard.*
- *April 20, 2004: S/MIME AES Requirement for SIP approved as Proposed Standard.*
- *April 13, 2004: RFC 3275 (Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP))*
- *March 31, 2004: RFC 3680 (A Session Initiation Protocol (SIP) Event Package for Registrations)*
- *March 22, 2004: The Session Inititation Protocol (SIP) 'Replaces' Header approved as a Proposed Standard.*
- *February 19, 2004: RFC 3702 (Authentication, Authorization, and Accounting Requirements for the Session Initiation Protocol (SIP))*
- *January 7, 2004: Indicating User Agent Capabilities in the Session Initiation Protocol (SIP) approved as Proposed Standard.*
- *January 5, 2004: RFC 3666 (Session Initiation Protocol (SIP) Public Switched Telephone Network (PSTN) Call Flows)*
- *January 5, 2004: RFC 3665 (Session Initiation Protocol (SIP) Basic Call Flow Examples)*
- *October 10, 2003: RFC 3605 (Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP))*
- *October 10, 2003: RFC 3603 (Private Session Initiation Protocol (SIP) Proxy-to-Proxy Extensions for Supporting the PacketCable Distributed Call Signaling Architecture)*
- *October 8, 2003: RFC 3806 (Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration) published*
- *August 26, 2003: RFC 3581 (An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing) published*
- *August 1, 2003: Session Initiation Protocol Extension Header Field for Service Route Discovery During Registration approved as a Proposed Standard*
- *July 30, 2003: RFC 3319 (Dynamic Host Configuration Protocol (DHCPv6) Options for Session Initiation Protocol (SIP) Servers) published*
- *April 28, 2003: RFC 3524 (Mapping of Media Streams to Resource Reservation Flows) published*
- *April 23, 2003: A Presence Event Package for the Session Initiation Protocol (SIP)', 'A Session Initiation Protocol (SIP) Event Template-Package for Watcher Information' and 'An Extensible Markup Language (XML) Based Format for Watcher Information' approved as Proposed Standards.*
- *April 10, 2003: RFC 3515 (The Session Initiation Protocol (SIP) Refer Method) published*
- *New IPC software tool for displaying SIP call flows.*
- *March 4, 2003: RFC 3485 (The Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Static Dictionary for Signaling Compression (SigComp)) published*
- *March 4, 2003: RFC 3487 (Requirements for Resource Priority Mechanisms for the Session Initiation Protocol (SIP)) published*

- *March 4, 2003:* RFC 3486 *(Compressing the Session Initiation Protocol (SIP)) published*
- *January 23, 2003:* RFC 3313 *(Private Session Initiation Protocol (SIP) Extensions for Media Authorization) published*
- *January 13, 2003:* 'Requirements for Resource Priority Mechanisms for the Session Initiation Protocol' *approved as an Informational RFC.*
- *December 19, 2002:* RFC 3388 *(Grouping of Media Lines in Session Description Protocol (SDP)) published*
- *December 17, 2002:* 'Mapping of Media Streams to Resource Reservation Flows' *approved as Proposed Standard*
- *December 16, 2002:* 'The Refer Method' *approved as Proposed Standard*
- *December 16, 2002:* 'A Session Initiation Protocol (SIP) Event Package for Registrations' *approved as Proposed Standard*
- *December 13, 2002:* RFC 3327 *(Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts) published*
- *December 6, 2002:* RFC 3398 *(Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping) published*
- *December 6, 2002:* RFC 3326 *(The Reason Header Field for the Session Initiation Protocol (SIP)) published*
- *December 6, 2002:* RFC 3428 *(Session Initiation Protocol (SIP) Extension for Instant Messaging) published*
- *December 4, 2002:* RFC 3323 *(A Privacy Mechanism for the Session Initiation Protocol (SIP)) published*
- *December 4, 2002:* RFC 3325 *(Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks) published*
- *November 26, 2002:* 'The Session Initiation Protocol (SIP) and Session Description Protocol (SDP) static dictionary for Signaling Compression (SigComp)' and 'Compressing the Session Initiation Protocol' *approved as Proposed Standards.*
- *November 7, 2002:* 'DHCPv6 Options for SIP Servers' *approved as Proposed Standard.*
- *November 5, 2002: PDF versions of the core SIP specifications are now* available.
- *November 1, 2002:* CPL schema definition
- *September 23, 2002:* 'Internet Media Types message/sipfrag' *approved as Proposed Standard.*
- *September 20, 2002:* 'Session Initiation Protocol Extension for Instant Messaging' *approved as a Proposed Standard.*
- *September 2002:* RFC 3311 *(The Session Initiation Protocol (SIP) UPDATE Method) published*
- *September 2002:* RFC 3372 *(Session Initiation Protocol for Telephones (SIP-T): Context and Architectures) published*
- *September 1, 2002:* IETF liaison statement to 3GPP
- *August 28, 2002:* SDP Simple Capability Declaration *approved as Proposed Standard.*
- *August 27, 2002:* RFC 3361 *(Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers) published*
- *August 26, 2002:* Grouping of m lines in SDP *approved as Proposed Standard.*
- *August 15, 2002:* RFC 3351 *(User Requirements for the Session Initiation Protocol (SIP) in Support of Deaf, Hard of Hearing and Speech-impaired Individual) published*
- *July 22, 2002:* List *of differences between draft-ietf-sip-rfc2543bis-09 and RFC 3261.*
- *July 9, 2002: 'SIP for Telephones (SIP-T): Context and Architectures' (draft-ietf-sipping-sipt-04.txt)* approved *as a BCP.*
- *July 3, 2002:* RFC 3261 *(SIP: Session Initiation Protocol),* RFC 3262 *(Reliability of Provisional Responses in Session Initiation Protocol (SIP)),* RFC 3263 *(Session Initiation Protocol (SIP): Locating SIP Servers),* RFC 3264 *(An Offer/Answer Model with Session Description Protocol (SDP)),* RFC 3265 *(Session Initiation Protocol (SIP)-Specific Event Notification),* RFC 3266 *(Support for IPv6 in Session Description Protocol (SDP)) published*

- *June 24, 2002: Signaling Compression approved as a Proposed Standard; SigComp - Extended Operations and Signaling Compression Requirements & Assumptions approved for publication as Informational RFCs.*
- *June 24, 2002: Extensions to the Session Initiation Protocol (SIP) for Asserted Identity and Short term requirements for Network Asserted Identity approved for publication as Informational RFCs*
- *June 24, 2002: Reason Header Field for the Session Initiation Protocol and SIP Extension for Registering Non-Adjacent Contacts approved as Proposed Standards.*
- *May 17, 2002: SIP UPDATE and resource management approved as Proposed Standards.*
- *May 6/7, 2002: SIP/SIPPING interim meeting in Las Vegas.*
- *April 6, 2002: SIP DHCP has been approved as Proposed Standard.*
- *March 7, 2002: SIP "bis" and related documents have been approved as Proposed Standards.*
- *February 22, 2002: The revised SIP specification, along with Reliability of Provisional Responses in SIP, SIP: Locating SIP Servers, SIP-Specific Event Notification, and An Offer/Answer Model with SDP, have been submitted for IETF last call.*
- *February 5, 2002: CPL was approved as a Proposed Standard.*
- *January 18, 2002: RFC 3219 (Telephony Routing over IP (TRIP)) published*
- *The tenth SIP interoperability test event will be held April 22-26, 2002 in Cannes, France at the Royal Casino Hotel, hosted by ETSI.*
- *December 2001: The ninth SIP interoperability test event took place in San Diego, hosted by Nuera.*
- *October 26, 2001: A completely editorially revised version of the SIP specification has been released for comment.*
- *The eighth SIP interoperability test event took place August 13-17, 2001 in Cardiff, UK.*
- *July 25, 2001: AOL submits statement on use of SIMPLE to FCC*
- *June 29, 2001: picture gallery of some SIP products and icons for SIP servers*
- *May 29, 2001: RFC2543bis (-03) draft*
- *May 4, 2001: Information about the 8th SIP Interoperability Test Event is now available*
- *April 14, 2001: search feature added.*
- *April 11, 2001: The SIP interoperability test event has a new logo, courtesy of Ubiquity.*

  SiPit
  SIP interoperability test event
- *April 10, 2001: RFC 3087 (Control of Service Context using SIP Request-URI) published*
- *Feb. 1, 2001: RFC 3050 (Common Gateway Interface for SIP) (sip-cgi) published*
- *Nov. 30, 2000: Caller preferences draft in WG last call until December 24, 2000*
- *Nov. 29, 2000: Guidelines for Authors of SIP Extensions draft in WG last call until December 24, 2000*
- *November 24, 2000: RFC2543bis (-02) draft*
- *Nov. 17, 2000: CPL in IESG last call.*
- *RFC 2976 (The SIP INFO Method) published*
- *The sixth SIP interoperability test event took place December 5-8, 2000 at Sylantro and Sun in Silicon Valley, California.*
- *June 20, 2000: The SIP Forum was founded. "SIP Forum is a non profit association whose mission is to promote awareness and provide information about the benefits and capabilities that are enabled by SIP."*
- *The fifth SIP interoperability test event took place August 8-10, 2000 at pulver.com in Melville, Long Island.*
- *June 15, 2000: RFC 2848, The PINT Service Protocol: Extensions to SIP and SDP for IP Access to Telephone Call Services, published.*
- *Added SIP internship and job listing.*
- *The fourth SIP interoperability test event took place April 17-19, 2000 in Rolling Meadows (near*

- *Chicago), Illinois, hosted by 3Com.*
- *February 28, 2000: Draft The SIP INFO Method is in IETF last call for Proposed Standard.*
- *September 1999: A new IETF working group on SIP has been created.*
- *The third SIP interoperability test event took place December 6th through 8th, 1999 in Richardson, Texas, hosted by Ericsson.*
- *The second SIP interoperability test event took place August 5th and 6th, 1999 at pulver.com (Melville, NY).*
- *The first SIP interoperability test event (known as "bake off") took place April 8th and 9th, 1999 at Columbia University, New York.*
- *SIP is a Proposed Standard (Feb. 2, 1999) published as RFC 2543 (March 17, 1999).*
- *New list of public SIP servers.*

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SIP Overview

SIP provides the necessary protocol mechanisms so that end systems and proxy servers can provide services:

- call forwarding, including
  - the equivalent of 700-, 800- and 900- type calls;
  - call-forwarding no answer;
  - call-forwarding busy;
  - call-forwarding unconditional;
  - other address-translation services;
- callee and calling ``number" delivery, where numbers can be any (preferably unique) naming scheme;
- *personal mobility*, i.e., the ability to reach a called party under a single, location-independent address even when the user changes terminals;
- terminal-type negotiation and selection: a caller can be given a choice how to reach the party, e.g., via Internet telephony, mobile phone, an answering service, etc.;
- terminal capability negotiation;
- caller and callee authentication;
- blind and supervised call transfer;
- invitations to multicast conferences.

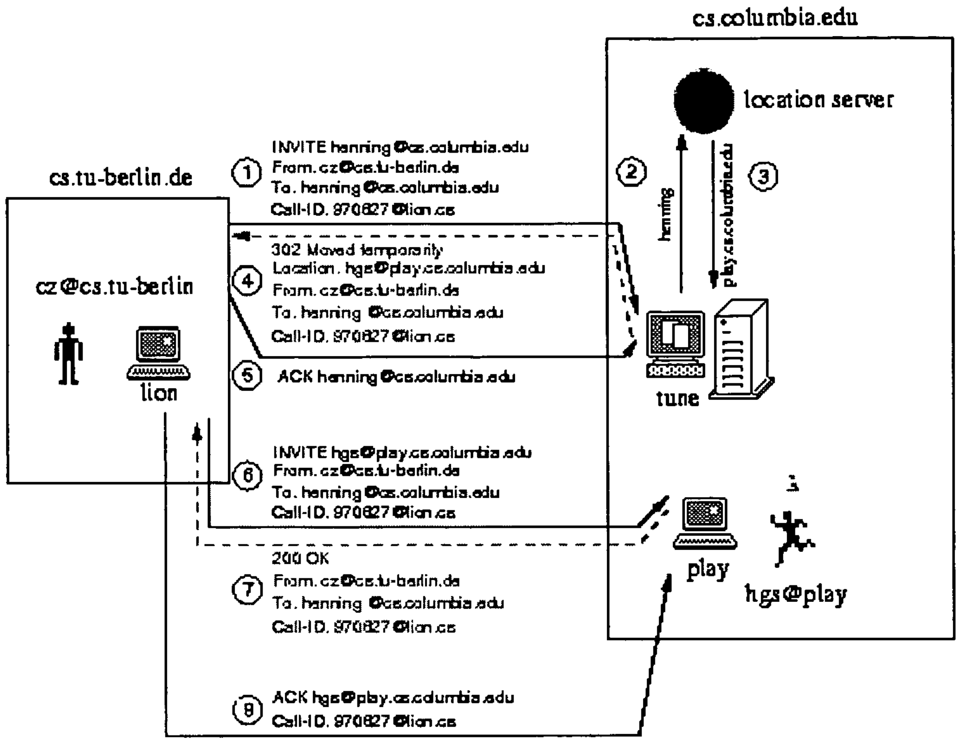
Extensions of SIP to allow third-party signaling (e.g., for click-to-dial services, fully meshed conferences and connections to multipoint control units (MCUs), as well as mixed modes and the transition between those) are available.

SIP addresses users by an email-like address and re-uses some of the infrastructure of electronic mail delivery such as DNS MX records or using SMTP EXPN for address expansion. SIP addresses (URLs) can also be embedded in web pages. SIP is addressing-neutral, with addresses expressed as URLs of various types such as SIP, H.323 or telephone (E.164).

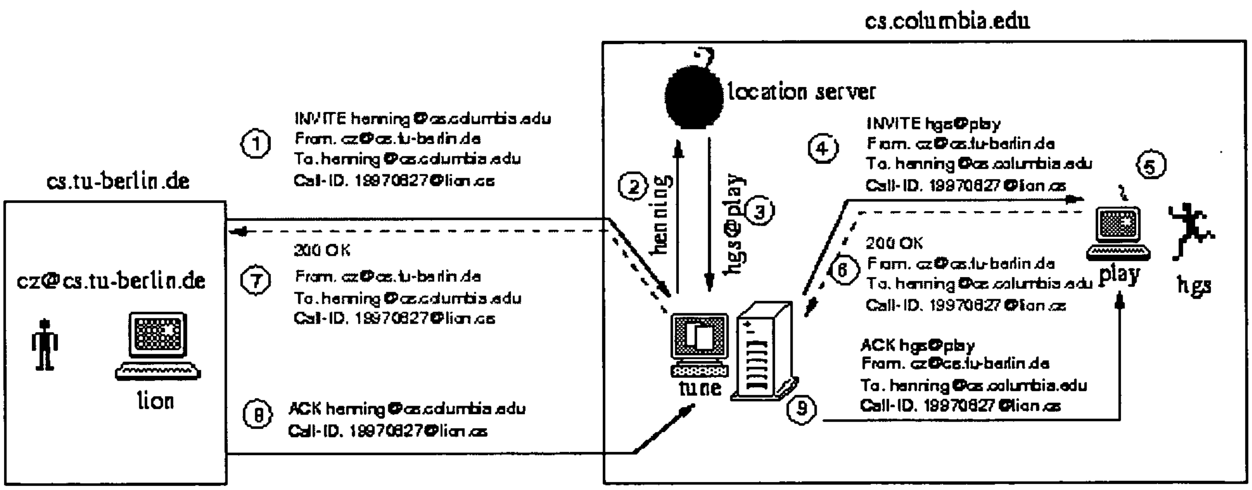
SIP can also be used for signaling Internet real-time fax delivery. This requires no major changes. Fax might be carried via RTP, TCP (e.g., the protocols discussed in the Internet fax WG) or other mechanisms.

SIP is independent of the packet layer and only requires an unreliable datagram service, as it provides its own reliability mechanism. While SIP typically is used over UDP or TCP, it could, without technical changes, be run over IPX, or carrier pigeons, frame relay, ATM AAL5 or X.25, in rough order of desireability.

SIP Operation in Redirect Mode



SIP Operation in Proxy Mode



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SIP-Related Work

The SIP WG has primary responsibility for the future development of SIP, but SIP-related work occurs in a number of IETF working groups:

SIP WG

> SIP standardization has moved from MMUSIC to the SIP working group, with its own mailing list and additional information, such as tasks, schedules and design teams.

SIPPING (Session Initiation Protocol Project INvestiGation)

> "working group is chartered to document the application of SIP to certain domain tasks and to develop requirements for the changes or extensions to SIP needed to accomplish those tasks. The SIPPING working group will concentrate on the frameworks, requirements, and practices related to SIP and its extensions, and will not specify changes or extensions to SIP."
>
> The supplemental web site contains schedules, working group drafts and other organizational information.

MMUSIC WG

> For SDP, SAP and related topics.

3GPP

> Release 5 and later use SIP in its Internet Multimedia Subsystem (IMS). The current status list is a good summary of the drafts, while specs/latest contains the most current drafts. In particular, TS 23.228 (architecture), 24.228 (call flows) and TS 29.328 contain most of the SIP-related material.

PINT

> PINT uses a profile of SIP.

iptel WG

> The Call Processing Language is designed to work with SIP and control its operation. The Gateway Location Protocol can be used to introduce proxy servers to SIP.

impp WG (Instant Messaging and Presence Protocol)

> SIP has been proposed as a solution, one of several, to the Internet presence and notification problem.

ETSI-Tiphon: DTS/TIPHON-03018 - Interface Protocol Requirements Definition; Implementation of TIPHON architecture using SIP

Non-Standardization Groups

- **SIP for mobility web pages**

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SIP Extensions

Now that the basic SIP specification has been completed, The MMUSIC working group aims to create a set of building blocks that build, incrementally, on this specification, making it useful for more specialized applications or satisfying additional domain-specific requirements. In my view, the ideal case is that these can be decomposed, where each building block is useful for more than one application or solves more than one problem. So far, we've identified these areas:

- reliable provisional responses;
- pre-call media cut-through ("183");
- SIP methods for mid-call SIP transactions that do not change SIP state;
- SDP QOS extensions and interaction of resource reservations with SIP;
- call control such as call transfer and MCUs;
- caller preferences;
- multipart message bodies for carrying non-SIP signaling messages;

Beyond these specific extensions, guidelines for mapping other signaling protocols, such as Q.931, ISUP and H.323, into SIP need to be developed. These guidelines would not add any components to SIP and the extensions, but rather provide information to implementors on how to combine the various elements to achieve the desired functionality.

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SIP RFCs and Drafts

Archive
- Archive for drafts

Mailing list archive
- HTML, through March 1998

All SIP-related drafts
- http://www.cs.columbia.edu/sip/drafts/

SIP-related RFCs:

RFC 3524	*Mapping of Media Streams to Resource Reservation Flows*	This document defines an extension to the Session Description Protocol (SDP) grouping framework. It allows requesting a group of media streams to be mapped into a single resource reservation flow. The SDP syntax needed is defined, as well as a new "semantics" attribute called Single Reservation Flow (SRF).
RFC 3515	*The Session Initiation Protocol (SIP) Refer Method*	Defines the REFER method. This Session Initiation Protocol (SIP) extension requests that the recipient REFER to a resource provided in the request. It provides a mechanism allowing the party sending the REFER to be notified of the outcome of the referenced request. This can be used to enable many applications, including call transfer. In addition to the REFER method, this document defines the the refer event package and the Refer-To request header.
RFC 3487	*Requirements for Resource Priority Mechanisms for the Session Initiation Protocol (SIP)*	Summarizes requirements for prioritizing access to circuit-switched network, end system and proxy resources for emergency preparedness communications using the Session Initiation Protocol (SIP).
RFC 3486	*Compressing the Session Initiation Protocol (SIP)*	Describes a mechanism to signal that compression is desired for one or more Session Initiation Protocol (SIP) messages. It also states when it is appropriate to send compressed SIP messages to a SIP entity.
RFC 3485	*The Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Static Dictionary for Signaling Compression (SigComp)*	The Session Initiation Protocol (SIP) is a text-based protocol for initiating and managing communication sessions. The protocol can be compressed by using Signaling Compression (SigComp). Similarly, the Session Description Protocol (SDP) is a text-based protocol intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation. This memo defines the SIP/SDP-specific static dictionary that SigComp may use in order to achieve higher higher efficiency. The dictionary is compression algorithm independent.
RFC 3428	*Session Initiation Protocol (SIP) Extension for Instant*	Instant Messaging (IM) refers to the transfer of messages between users in near real-time. These messages are usually, but not required to be, short. IMs are often used in

	Messaging	a conversational mode, that is, the transfer of messages back and forth is fast enough for participants to maintain an interactive conversation. This document proposes the MESSAGE method, an extension to the Session Initiation Protocol (SIP) that allows the transfer of Instant Messages. Since the MESSAGE request is an extension to SIP, it inherits all the request routing and security features of that protocol. MESSAGE requests carry the content in the form of MIME body parts. MESSAGE requests do not themselves initiate a SIP dialog; under normal usage each Instant Message stands alone, much like pager messages. MESSAGE requests may be sent in the context of a dialog initiated by some other SIP request.
RFC 3420	*Internet Media Type message/sipfrag*	This document registers the message/sipfrag Multipurpose Internet Mail Extensions (MIME) media type. This type is similar to message/sip, but allows certain subsets of well formed Session Initiation Protocol (SIP) messages to be represented instead of requiring a complete SIP message. In addition to end-to-end security uses, message/sipfrag is used with the REFER method to convey information about the status of a referenced request.
RFC 3388	*Grouping of Media Lines in Session Description Protocol (SDP)*	Extensions to SDP that allow grouping of media streams for lip synchronization and to represent the same content on different network addresses
RFC 3361	*Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers*	Defines a DHCP option for locating the outbound SIP proxy server
RFC 3319	*Dynamic Host Configuration Protocol (DHCPv6) Options for Session Initiation Protocol (SIP) Servers*	Defines a DHCPv6 options for locating the outbound SIP proxy server
RFC 3327	*Session Initiation Protocol (SIP) Extension Header Field for Registering Non-Adjacent Contacts*	Defines the `Path` header field that registers a list of proxies between the UA and the registrar
RFC 3326	*The Reason Header Field for the Session Initiation Protocol (SIP)*	For creating services, it is often useful to know why a Session Initiation Protocol (SIP) request was issued. This document defines a header field, Reason, that provides this information. The Reason header field is also intended to be used to encapsulate a final status code in a provisional

		response. This functionality is needed to resolve the "Heterogeneous Error Response Forking Problem", or HERFP.
RFC 3325	*Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks*	Defines **P-Asserted-Identity** and **P-Preferred-Identity** header fields, allowing SIP proxies to add user identity information and callers to request privacy
RFC 3324	*Short Term Requirements for Network Asserted Identity*	Defines requirements for caller identities established by network entities
RFC 3323	*A Privacy Mechanism for the Session Initiation Protocol (SIP)*	Describes SIP caller privacy issues and defines the **Privacy** header field
RFC 3329	*Security Mechanism Agreement for the Session Initiation Protocol (SIP)*	This document defines new functionality for negotiating the security mechanisms used between a Session Initiation Protocol (SIP) user agent and its next-hop SIP entity. This new functionality supplements the existing methods of choosing security mechanisms between SIP entities.
RFC 3313	*Private Session Initiation Protocol (SIP) Extensions for Media Authorization*	Describes the need for Quality of Service (QoS) and media authorization and defines a Session Initiation Protocol (SIP) extension that can be used to integrate QoS admission control with call signaling and help guard against denial of service attacks. The use of this extension is only applicable in administrative domains, or among federations of administrative domains with previously agreed-upon policies, where both the SIP proxy authorizing the QoS, and the policy control of the underlying network providing the QoS, belong to that administrative domain or federation of domains.
RFC 3312	*Integration of Resource Management and SIP*	Framework for preconditions
RFC 3311	*The Session Initiation Protocol (SIP) UPDATE Method*	This specification defines the new UPDATE method for the Session Initiation Protocol (SIP). UPDATE allows a client to update parameters of a session (such as the set of media streams and their codecs) but has no impact on the state of a dialog. In that sense, it is like a re-INVITE, but unlike re-INVITE, it can be sent before the initial INVITE has been completed. This makes it very useful for updating session parameters within early dialogs.
RFC 3261 (Bookmarks kindly	*SIP: Session Initiation Protocol*	Core protocol specification; obsoletes RFC 2543.

provided by Alexandre Gilles)		
RFC 3262	*Reliability of Provisional Responses in the Session Initiation Protocol (SIP)*	Making 1xx responses reliable; introduces PRACK method
RFC 3263	*Session Initiation Protocol (SIP): Locating SIP Servers*	Describes DNS mechanisms (NAPTR, SRV) for locating SIP servers
RFC 3264	*An Offer/Answer Model with the Session Description Protocol (SDP)*	How SDP is used within SIP to negotiate sessions
RFC 3265	*Session Initiation Protocol (SIP)-Specific Event Notification*	SIP event model; defines SUBSCRIBE and NOTIFY
RFC 3087	*Control of Service Context using SIP Request-URI*	Defines how the SIP URI can be used to invoke services such as voicemail
RFC 3050	*Common Gateway Interface for SIP*	sip-cgi, as scripting interface
RFC 2976	*The SIP INFO Method*	Defines INFO method for carrying SIP-related information
RFC 2848	*The PINT Service Protocol: Extensions to SIP and SDP for IP Access to Telephone Call Services*	Defines how SIP events can be used to invoke PSTN services such as Internet call waiting

Summary of SIP-Related Standardization Efforts

There are a number of extensions for adding features to SIP. Current drafts are listed below. Only drafts whose names start with `draft-ietf-sip-` and `draft-ietf-sipping-` are SIP (or SIPPING) working group work items, while others are individual submissions by their authors. Individual submissions may later become working group items. A draft does not have to be labeled as a WG item to be progressed.

Given the number of drafts, it may be hard to track which drafts are significant for different applications. Below, some of the efforts are summarized and pointers are provided to current drafts. All of these efforts are believed to be active. Some of the text below was contributed by Jonathan Rosenberg.

Core SIP Specification

The core SIP specification is RFC 3261, which obsoletes RFC 2543. Differences to the last Internet Draft version, bis09, are are minor. Related specifications are: RFC 3262 (*Reliability of Provisional*

Responses in Session Initiation Protocol (SIP)), RFC 3263 (*Session Initiation Protocol (SIP): Locating SIP Servers*), RFC 3264 (*An Offer/Answer Model with Session Description Protocol (SDP)*), RFC 3265 (*Session Initiation Protocol (SIP)-Specific Event Notification*), and RFC 3266 (*Support for IPv6 in Session Description Protocol (SDP)*).

Informational Documents

Call Flows

To help implementors, a set of call flows has been published that give examples of common call setup and registration scenarios.

Guidelines for Writing SIP Extensions

A number of extensions are being proposed to add headers or methods to SIP. Extensions should follow a set of rules to maximize the chance that different extensions can coexist. The draft proposes criteria for evaluating what is and what is not a good SIP extension, and describes things all extensions need to discuss. It will eventually become a BCP RFC. This is a SIP WG work item.

SIP Through NATs and Firewalls

These drafts presents information on issues that arise in getting SIP and related multimedia services through NATs and firewalls. The work will result in an informational RFC.

SIP Enabled Services to Support the Hearing Impaired

This document outlines a set of services enabled by SIP that allow for access to voice services by people who are hearing impaired.

User Agent Configuration

SIP user agents need to determine whether to use an outbound proxy and where to send registration updates. The address of the outbound proxy can be configured manually and the registration can be sent via multicast. DHCP is an additional method for configuring this information. DHCP is used extensively to configure boot-time information in IP-connected hosts. (This is a SIP WG work item. It is currently in IESG review.)

For more sophisticated selection of proxies, the Service Location Protocol allows proxies and registrars to advertise their capabilities. In large networks, users may have a choice about the SIP server they connect to. Different servers can provide different services to their users; for example, some may support CPL execution, and others may not. Some may support IPSec, and some may not. This work defines a way in which SIP end systems can discover SIP servers providing specific capabilities. It is done through the Service Location Protocol (SLP), specified in RFC2608. To enable SIP server discovery with SLP, a template needs to be defined which basically defines the schema for SIP servers. The work will be turned into an IANA registration, as per normal SLP procedures. This will allow existing SLP servers to provide SIP server discovery.

Network Management

A SIP SNMP MIB is under development. It provides monitoring of SIP message processing,

configuration, and alarms for SIP enabled user agents, gateways, and proxies. The work is preliminary and scheduled to complete towards the end of 2000.

Infrastructure Improvements

Reliable Provisional Responses

In the base SIP specification, provisional responses (100 through 199, also called informational) are transmitted on a best-effort basis, i.e., without guarantee that the client will receive a particular response. However, applications, such as PSTN interworking and for establishing preconditions on call establishment, have arisen which derive state transitions from these messages. A new method, PRACK has been defined to allow clients to request that provisional responses are retransmitted by the server until received by the client. This is a SIP WG work item and nearly complete.

Supported Header

SIP allows for extensions whereby the client can mandate that the server understand the extension in order to process the response. However, there is no way for the server to determine what features are supported by the client, so that the server can use those features in a response. A new header called Supported provides this feature. The work is complete and currently under IESG review.

Session Timer

SIP sessions exist until either side tears them down. Some proxies and end systems, however, would like to be assured that both sides are still alive and interested in participating. A session timer extension provides a simple keep-alive, based on the soft-state refresh principle. This is a SIP WG work item and nearly complete.

Service Enablers

Several drafts are proposing new service enabling capabilities to SIP. These are not services themselves, but primitives that enable groups of services.

Third-Party Call Control

A SIP user agent can set up calls between two other SIP user agents. This third-party call control makes it easy to provide services such as click-to-dial, mid-call announcements and conferences. This service does not require any protocol additions.

Caller Preferences

Calls are typically routed by the callee's proxies and end systems. The caller preferences extension describes how the caller can indicate its preferences as to how requests should be handled. These advanced routing services are enabled through the addition of new parameters that can be registered by call recipients. This is a SIP WG work item and has been stable for quite some time.

Preconditions

In some cases, it is necessary to delay ringing the callee until a set of preconditions has been established. One examle is reservation-based quality-of-service. There, a separate resource reservation protocol

determines whether sufficient network resources are available. Only when this step succeeds should the phone ring. Similarly, ringing may be conditional on being able to set up a secure media session. This extension adds preconditions to SDP, so that session establishment can be made conditional to QoS or security establishment. It adds a new SIP method, COMET, to confirm that conditions have been met. This is a SIP WG work item that is considered stable.

Call Transfer

Services such as call transfer require additional methods. This extension allows one entity to request another to make a call. It enables call transfer, in addition to other services such as conferencing. This is a SIP WG work item. The basic procedure and syntax is considered stable.

PSTN Interconnection

SIP-based systems are often going to be connected to PSTN gateways. For analog and ISDN circuits, no particular additions are required, but interconnection with Signaling System #7 (SS7) requires additional protocol support to transparently interwork and bridge. Among other tasks, this set of drafts describe a new SIP method, INFO, to carry non-call-state-changing mid-call ISUP and QSIG messages across a SIP cloud as MIME attachments. There is also a draft that describes how ISUP messages and SIP messages relate to each other at a gateway.

The PSTN uses pre-call tones and announcements to provide information on call progress. This draft proposes a new response code, 183, and a simple mechanism to enable them in SIP. The work is largely complete.

Internet telephony needs to provide emergency calling, i.e., 911 calling.

Quality of Service

SIP does not reserve network resources, but can be helpful in providing. authorization, authentication and accounting (AAA), possibly in conjunction with the Open Settlement Protocol.

Presence, Events and Instant Messaging

SIP can also be easily extended to support presence and instant messaging. Voice mail notification, also known as message waiting service, is an example of using the basic event mechanism to provide a classical telephony service.

This extension may also make SIP suitable to control home appliances.

Creating SIP Services

A number of APIs and other mechanisms to provide SIP-related services have been developed, including an interface similar to the web common gateway interface (CGI), called sip-cgi, a programming language, CPL, for a limited set of features in proxies and Java servlets. In addition, there are APIs, such as JAIN and Parlay, that provide programmable services.

Servers can be configured with CPL and SIP-CGI scripts in a variety of ways. One mechanism is to upload the scripts via ">REGISTER requests from SIP user agents.

Mobile Hosts

When SIP UAs visit networks, they have a number of choices of how to interact with the local network, if at all, and where to register. Such interaction may be necessary due to firewalls and authentication.

Other Efforts

Below is an incomplete list of other SIP standardization efforts; none of them are part of the SIP WG charter.

- Alignment with the PacketCable DCS specifications;
- Exploration of applications to SIP in mobile environments;
- Enabling SIP to emulate residential phone service
- Interworking of SIP state machines and IN call models;
- Interworking with the MLPP multi-level priority specifications of military ISUP;
- Integration of QoS, policy, and signaling;
- Relationship between SIP, QoS and OSP as well as AAA architecture;
- SIP for selecting MPLS routes.

Current Documents

- Extensions of the basic specification to handle reliable 1xx messages, call flows, call control;
- SIP for presence;
- SIP extensions for device control;
- Using DHCP and SLP to configure SIP servers and clients
- Making SIP work through firewalls and NATs
- Distributed Call Signaling, a set of PacketCable proposals;
- Mobility-related;
- SIP APIs such as sip-cgi and servlets;
- Caller preferences allow callers to guide the behavior of proxies and user agents;
- QoS negotiation;
- PSTN interworking, such as carriage of ISUP messages, mid-call signaling;
- H.323 interworking;
- Security (authentication, encryption) issues;
- Of historical interest only.
- SIPstone: benchmarking SIP servers

Last updated 02/10/2005 13:52:58 by Henning Schulzrinne

Index of ABNF

These HTML versions of ABNF have been automatically generated using the **ABNF2HTML** tool provided by Inet Technologies, Inc. written by Brian Bidulock.

These versions of ABNF are for informational purposes only and are intended as a tool to help read the ABNF specifications. Please see the Copyright and Disclaimers at the end of each document. For definitive syntax descriptions, please refer to the source IETF documents.

The protocol syntax is presented in ABNF according to RFC2234 with extensions from RFC822.

The syntaxes presented are:

- draft-ietf-sip-rfc2543bis-00 from draft-ietf-sip-rfc2543bis-00.txt
- draft-ietf-sip-rfc2543bis-01 from draft-ietf-sip-rfc2543bis-01.txt

- rfc1806 from rfc1806.txt
- rfc2068 from rfc2068.txt
- rfc2234 from rfc2234.txt
- rfc2543 from rfc2543.txt
- rfc2806 from rfc2806.txt

SIP Implementations

News

- September 17, 2001: Tellme studio

Introduction

Below are some implementations in progress or completed. If no URL is shown for the product, the software or hardware is not yet available. Implementations are listed alphabetically by organization.

Not all of the implementations described below are commercial products. Please contact the individual companies and institutions for details. Descriptions were provided by the organization listed.

Many other sites list SIP implementations:

- Carl Lindner
- pulver.com
- voxilla.org
- iptel.org
- iptelephony collect information about IP telephony software.
- Cisco has a list of their SIP-enabled products.
- voip-info

A picture gallery of SIP hardware illustrates some of these.

Test Tools

See also a list of test-related documents such as PICS. sip_scenario can translate SIP call flows into pictures.

SIP Scenario SIp performance tester NastySip

Supporting Software

resparse for DNS SRV smaller SRV parser (Arnt Gulbrandsen, Vijay Gurbani) dnsjava; RULI (Resolver User Layer Interface)

Windows 2000 supports SRV records through its `DnsQuery()` interface.

Managed DNS for VoIP

SIP Libraries and Stacks

Stacks are written in C/C++ unless otherwise noted.

3Com AltiLogic Columbia University Columbia University (Java) Compaq Cosystems Technologies Ltd. Data Connection dynamicsoft (Java) HelloSoft HelloSIP Hughes Software Systems Indigo Software (Java) Ind Tele Soft Mediatrix Mistral Motorola NetBricks NIST OPAL oSIP RADvision RADvision Express reSIProcate RNID stack (Delphi 6) Telogy Trillium Ubiquity U4EA Vovida Vovida.org Wind River Systems

User Agents (Applications)

ActiveX-Control Alcatel Avaya IP phone Avaya softphone Avaz Columbia University sipc (P) CTL SIPHON Cornfed dissipate dynamicsoft eyeP PhoneeStara GMD Fokus Hotsip (P) Hughes Software Systems Indigo (P) Interactive Intelligence href="http://www.marconi.com/html/products/virtualpresencesystem.htm">Marconi (video) playSIP minisip SIPPstar tkcPhone (for Zaurus) KPhone (P) linphone Microppliances (soft client) Microsoft Portrait (video) for CE Motorola Pingtel Shtoom Siemens mySIP Siemens Siemens OpenScape SIPHON href="http://www.vovida.org/applications/downloads/sipset/">SIPset SIPquest SJ Labs Sonus Networks (PSX and IntelligentIP) swissvoice Ubiquity Vovida Wi-Fone for PocketPC xten XPhone for PocketPC

Implementations marked with (P) also support presence.

Ethernet Phones

3Com bcm Cisco (docs) E-tel Grandstream Pingtel Snom TuxScreen Way2Call Siemens (HiNet LP510) Siemens optiPoint Wylus

WiFi Phones

Zyxel Senao

PBX

Citel CTL Interactive Intelligence 8x8 Intraswitch Pingtel SIPquest

PSTN Gateways

8x8 Epygi Komodo Cisco ATA 186 IAD SIPURA SPA-2000 Interactive Intelligence Mediatrix Nuera Sonus Networks (GSX) T&S Software UCL Vegastream

Firewalls, NAT traversal and ALGs

Cisco Pix Entel ADSL modem Ingate Intertex IX66 Intertex IX 66 with ADSL modem Jasomi Microppliances Netrake

Proxies, Registrars and Redirect Servers

3Com Brekeke Cisco Columbia University sipd dynamicsoft Fomine Hewlett Packard Hotsip

Hughes Software Systems Indigo Interactive Intelligence iptel.org Meetinghouse Data Communications MicroAppliances ObjectSoftware partysip SIPfoundry sipX SIPquest Siptrex Snom Sonus Networks (PSX) T&S Software Terminal Technologies Ubiquity Vovida.org

3G Systems

BlueSlice Mobile SIP Exchange

Unified Messaging

Columbia University sipum CTL Interactive Intelligence Pingtel SIPquest

Conferencing and Other Media Servers

Columbia University sipconf CTL Hughes Software Systems Interactive Intelligence SIPquest Snom media server

SIP-H.323 Signaling Gateway

Columbia University siph323 Hughes Software Systems mySIP SIPquest Vovida VOCAL net.com ShoutIP

VoiceXML

Tellme Studio Pingtel SIPquest

Others

Organization	Name of software	Contact	Type	Description
Cisco	IOS 12.1(1)T		router OS	Cisco AS5300 access server, the Cisco 2600 and Cisco 3600 series routers.
ECI IP	Application solutions for SIP-enabled networks		Application services	enhanced calling services, settlement and clearinghouse services, broadband voice services, voice portal services, voice VPN services

CPL, SIP servlet and sip-cgi Implementations

Indigo Software has CPL implementations for both client and server sides (CPL "editor" and CPL "server") that can be coupled to a SIP UA and a SIP proxy, respectively. They are also working on an implementation of SIP-CGI.

SIP Servlet API Reference Implementation / Technology Compatibility Kit

Columbia University's sipd server handles sip-cgi; a CPL implementation is in progress. The sipc UA

can upload cgi and CPL scripts.

The dynamicsoft application server supports CGI, CPL, in addition to servlets. A white paper describes the motivation for this architecture.

Ubiquity's server supports CPL.

Institut für Kommunikationsnetze, TU-Wien is developing a service platform based on servlets.

Hughes Software Systems SIP Server provides a CPL engine. "To optimize performance, the SIP server converts raw XML into an internally optimized sequential execution format which results in significant performance gain over executing XML scripts for each service invocation."

Last updated 02/10/2005 13:53:21 by Henning Schulzrinne

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musi, 23:24 UTC, Mon 07 of Feb, 2005: i am making FINAL project on voip n vowip...any body having source code plz help me ...advane thanx...musi uet taxila (PAKISTAN)...enggsuleri@yahoo.com

money365, 01:01 UTC, Mon 07 of Feb, 2005: i am looking for a divice like call00.com if any1 can help me please email me on money365@gmail.com thx

josiahbryan, 01:41 UTC, Sun 06 of Feb, 2005: I havnt tried using AMP before - I prefer to do it by hand. Editing /etc/asterisk/extensions.conf is a lot easier than trying to install and wade through AMP. Just IMHO...

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- Silicon Chips specifically designed to support VOIP

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  - OSP, PacketCable

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  - IVR / Speech synthesis definition: SSML
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  - Telco Engineering Information

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- Suggestions: Put your requests and suggestions here
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- WikiNode

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Terms of Service Privacy Policy

RSS Wiki

[Execution time: 1.34 secs] [Memory usage: Unknown] [261 database queries used]
[GZIP Disabled] [Server load: 7.36]

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musi, 23:24 UTC, Mon 07 of Feb, 2005: i am making FINAL project on voip n vowip...any body having source code plz help me ...advane thanx...musi uet taxila (PAKISTAN)...enggsuleri@yahoo.com

money365, 01:01 UTC, Mon 07 of Feb, 2005: i am looking for a divice like call00.com if any1 can help me please email me on money365@gmail.com thx

josiahbryan, 01:41 UTC, Sun 06 of Feb, 2005: I havnt tried using AMP before - I prefer to do it by hand. Editing /etc/asterisk/extensions.conf is a lot easier than trying to install and wade through AMP. Just IMHO...

Read More...

Menu

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:: Wiki

- Network design and operation for VOIP
  - Bandwidth Requirements - How much bandwidth does VOIP use?
  - Billing - Billing VOIP users for service
  - Codecs for VOIP - converting analog voice signals to the digital domain
  - FAX and VOIP
  - How To Debug and Troubleshoot VOIP - Tools and Techniques for finding and fixing problems
  - Low Bandwidth Solutioins Using VOIP on dial-up or other low speed connections
  - NAT and VOIP NAT Traversal
  - QoS - Network Quality of Service for VOIP
  - Routers - Routers useful for VOIP
  - Session and Border Controllers and NAT traversal solutions
  - Software - Software for VOIP
  - Test Equipment: Analyze the speech quality of VoIP calls
  - VOIP over Satellite Using VOIP over satellite connections
  - Wireless VOIP: Cut the wires! Roam free with wireless VOIP

- VOIP Service Providers connecting users to each other and to the Public Switched Telephone Network
  - VOIP Service Providers
  - VOIP Service Providers Residential
  - VOIP Service Providers Business
  - VOIP Service Providers B2B

- Silicon Chips specifically designed to support VOIP

- VOIP Websites: Other VOIP websites on the Internet

- Policy and Regulatory: VOIP legal and regulatory information

- Protocols - the language of VOIP
  - IP Protocols: SIP, LTP, H.323, SCCP, MGCP, Megaco, IAX, STUN, ENUM, TRIP, SIMPI PINT, SCTP, T.37, T.38, COPS
  - ITU protocols: SS7, ISUP
  - ITU related standards: P.1010
  - OSP, PacketCable

- Markup Languages
  - IVR Presentation and dialog management: VoiceXML
  - Call control / conferencing / call routing: CCXML
  - IVR / Speech recognition definition: SRGS
  - IVR / Speech synthesis definition: SSML
  - IVR / prompting / recording / conferencing / DTMF / Voice: CallXML

- Organizations
  - Standards Organizations: IETF, ITU, CCITT, ANSI, ISO, IEC, ETSI, IEEE, W3C
  - VoIP User Groups: List of local VoIP User Groups
  - Industry: SIP Forum, H.323 Forum, SIPfoundry, ChinaVoIP

- Traditional Telephone Network
  - Analog Telephone Information
  - PBX features
  - Telecom terms: ANI, BRI, CALEA, CENTREX, Channelbank, CLEC, CPC, DID, DNIS, E ISDN, LATA, LNP, NEBS, NPA-NXX, Numbering plans, PBX, POTS, PRI, PSAP, PSTN, V and H Coordinates, 911, more
  - Telco Engineering Information

Suggestions and Questions

- How to add information to this wiki
- Suggestions: Put your requests and suggestions here
- Search VOIP sites on the Internet

Our wiki-Neighborhood

- WikiNode

Please update this page with new information (you'll have to login first), or without logging in, you c
a comment on this page -- click on the comment tab above. Thanks! - support@voip-info.org

Created by: system last modification: Thursday 10 of February, 2005 [16:36:05 UTC] by Joshua.Stephens

RSS Wiki

[Execution time: 1.34 secs] [Memory usage: Unknown] [261 database queries used]
[GZIP Disabled] [Server load: 7.36]

SIP Implementations

News

- September 17, 2001: Tellme studio

Introduction

Below are some implementations in progress or completed. If no URL is shown for the product, the software or hardware is not yet available. Implementations are listed alphabetically by organization.

Not all of the implementations described below are commercial products. Please contact the individual companies and institutions for details. Descriptions were provided by the organization listed.

Many other sites list SIP implementations:

- Carl Lindner
- pulver.com
- voxilla.org
- iptel.org
- iptelephony collect information about IP telephony software.
- Cisco has a list of their SIP-enabled products.
- voip-info

A picture gallery of SIP hardware illustrates some of these.

Test Tools

See also a list of test-related documents such as PICS. sip_scenario can translate SIP call flows into pictures.

SIP Scenario SIp performance tester NastySip

Supporting Software

resparse for DNS SRV smaller SRV parser (Arnt Gulbrandsen, Vijay Gurbani) dnsjava; RULI (Resolver User Layer Interface)

Windows 2000 supports SRV records through its `DnsQuery()` interface.

Managed DNS for VoIP

SIP Libraries and Stacks

Stacks are written in C/C++ unless otherwise noted.

3Com AltiLogic Columbia University Columbia University (Java) Compaq Cosystems Technologies Ltd. Data Connection dynamicsoft (Java) HelloSoft HelloSIP Hughes Software Systems Indigo Software (Java) Ind Tele Soft Mediatrix Mistral Motorola NetBricks NIST OPAL oSIP RADvision RADvision Express reSIProcate RNID stack (Delphi 6) Telogy Trillium Ubiquity U4EA Vovida Vovida.org Wind River Systems

User Agents (Applications)

ActiveX-Control Alcatel Avaya IP phone Avaya softphone Avaz Columbia University sipc (P) CTL SIPHON Cornfed dissipate dynamicsoft eyeP PhoneeStara GMD Fokus Hotsip (P) Hughes Software Systems Indigo (P) Interactive Intelligence href="http://www.marconi.com/html/products/virtualpresencesystem.htm">Marconi (video) playSIP minisip SIPPstar tkcPhone (for Zaurus) KPhone (P) linphone Microppliances (soft client) Microsoft Portrait (video) for CE Motorola Pingtel Shtoom Siemens mySIP Siemens Siemens OpenScape SIPHON href="http://www.vovida.org/applications/downloads/sipset/">SIPset SIPquest SJ Labs Sonus Networks (PSX and IntelligentIP) swissvoice Ubiquity Vovida Wi-Fone for PocketPC xten XPhone for PocketPC

Implementations marked with (P) also support presence.

Ethernet Phones

3Com bcm Cisco (docs) E-tel Grandstream Pingtel Snom TuxScreen Way2Call Siemens (HiNet LP510) Siemens optiPoint Wylus

WiFi Phones

Zyxel Senao

PBX

Citel CTL Interactive Intelligence 8x8 Intraswitch Pingtel SIPquest

PSTN Gateways

8x8 Epygi Komodo Cisco ATA 186 IAD SIPURA SPA-2000 Interactive Intelligence Mediatrix Nuera Sonus Networks (GSX) T&S Software UCL Vegastream

Firewalls, NAT traversal and ALGs

Cisco Pix Entel ADSL modem Ingate Intertex IX66 Intertex IX 66 with ADSL modem Jasomi Microppliances Netrake

Proxies, Registrars and Redirect Servers

3Com Brekeke Cisco Columbia University sipd dynamicsoft Fomine Hewlett Packard Hotsip

Hughes Software Systems Indigo Interactive Intelligence iptel.org Meetinghouse Data Communications MicroAppliances ObjectSoftware partysip SIPfoundry sipX SIPquest Siptrex Snom Sonus Networks (PSX) T&S Software Terminal Technologies Ubiquity Vovida.org

3G Systems

BlueSlice Mobile SIP Exchange

Unified Messaging

Columbia University sipum CTL Interactive Intelligence Pingtel SIPquest

Conferencing and Other Media Servers

Columbia University sipconf CTL Hughes Software Systems Interactive Intelligence SIPquest Snom media server

SIP-H.323 Signaling Gateway

Columbia University siph323 Hughes Software Systems mySIP SIPquest Vovida VOCAL net.com ShoutIP

VoiceXML

Tellme Studio Pingtel SIPquest

Others

Organization	Name of software	Contact	Type	Description
Cisco	IOS 12.1(1)T		router OS	Cisco AS5300 access server, the Cisco 2600 and Cisco 3600 series routers.
ECI IP	Application solutions for SIP-enabled networks		Application services	enhanced calling services, settlement and clearinghouse services, broadband voice services, voice portal services, voice VPN services

CPL, SIP servlet and sip-cgi Implementations

Indigo Software has CPL implementations for both client and server sides (CPL "editor" and CPL "server") that can be coupled to a SIP UA and a SIP proxy, respectively. They are also working on an implementation of SIP-CGI.

SIP Servlet API Reference Implementation / Technology Compatibility Kit

Columbia University's sipd server handles sip-cgi; a CPL implementation is in progress. The sipc UA

can upload cgi and CPL scripts.

The dynamicsoft application server supports CGI, CPL, in addition to servlets. A white paper describes the motivation for this architecture.

Ubiquity's server supports CPL.

Institut für Kommunikationsnetze, TU-Wien is developing a service platform based on servlets.

Hughes Software Systems SIP Server provides a CPL engine. "To optimize performance, the SIP server converts raw XML into an internally optimized sequential execution format which results in significant performance gain over executing XML scripts for each service invocation."

Last updated 02/10/2005 13:54:50 by Henning Schulzrinne